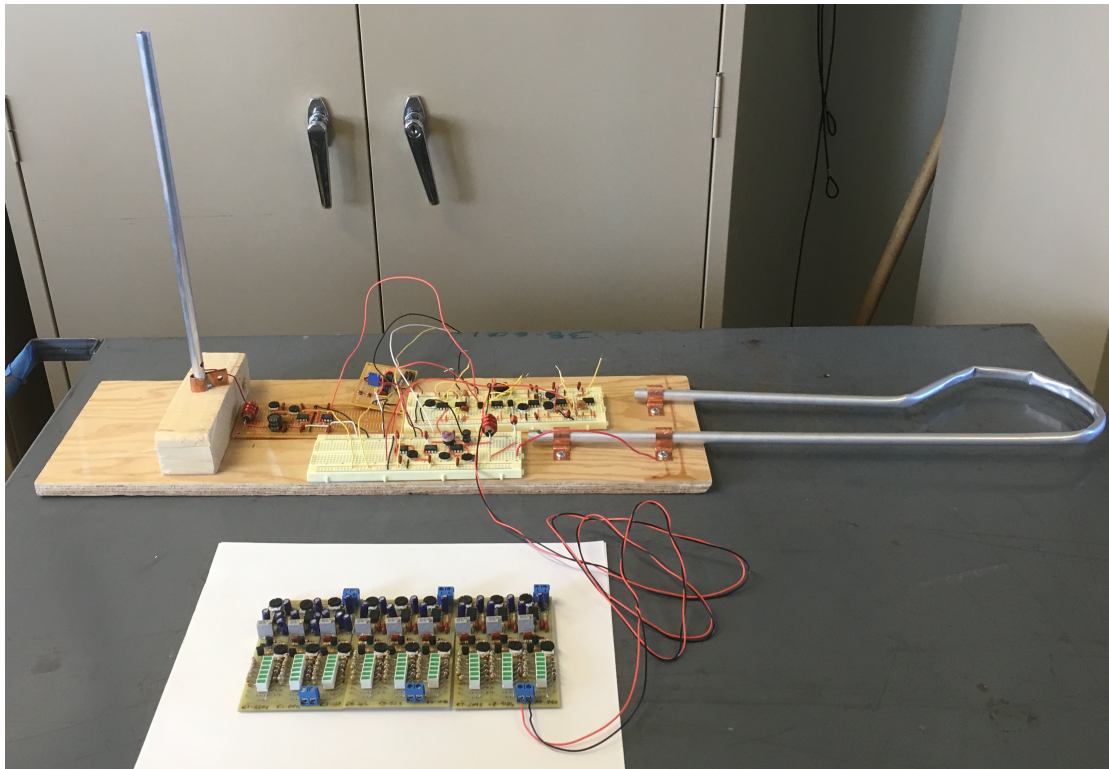


6.101 Final Project

Theremin

Pedro Brito · David Gomez · Patrick McCabe

May 12, 2016



Abstract

The goal of this project is to create a theremin. A theremin is a musical instrument that is played without physical contact. This is achieved by using two antennas, one controls the pitch and the other the volume, which are capacitively coupled to the player's hands. By moving their hands around these antennas, the player can change the capacitance of the coupling. This change in capacitance is then detected allowing a tone that varies in pitch and volume based on distance from the antennas to be produced.

Playing a theremin, specifically producing a desired pitch, is difficult because there is no physical reference point to know where a hand should be moved. To aid in this, a pitch visualizer has also been included in this project. This visualizer has several LED indicators that illuminate when the theremin plays certain musical notes to give an easy reference to the player.

Contents

1	Motivation	5
2	System Architecture	5
2.1	High Level	5
2.2	Pitch Oscillators (David)	6
2.2.1	Purpose	6
2.2.2	Theory - Sine Wave Oscillator	6
2.2.3	Theory - Detecting Capacitance Change	7
2.2.4	Circuit - Variable Pitch Oscillator	9
2.2.5	Theory - Fixed Frequency Oscillator	10
2.2.6	Circuit - Fixed Pitch Oscillator	11
2.3	Volume Oscillator (David)	13
2.3.1	Purpose	13
2.3.2	Design - Volume Oscillator	13
2.3.3	Circuit - Volume Oscillator	13
2.4	Audio Generation (Patrick)	14
2.4.1	Purpose	14
2.4.2	Theory - Signal Mixing	14
2.4.3	Circuit - Signal Mixing	15
2.4.4	Theory - Frequency to DC Voltage Conversion	16
2.4.5	Circuit - Frequency to DC Voltage Conversion	16
2.4.6	Theory - Voltage Controlled Attenuator	18
2.4.7	Circuit - Voltage Controlled Attenuator	18
2.5	Frequency Visualization (Pedro)	19
2.5.1	Purpose	19
2.5.2	Design	20
2.5.3	Circuit - Gain Stage	20
2.5.4	Circuit - Band-Pass Filtering Stage	21
2.5.5	Circuit - Amplitude Detection Stage	22
2.5.6	Circuit - LED Visualization Stage	23
2.5.7	Visualizer Circuit Schematic and PCB	23
3	Integration	26
4	Conclusion	30

List of Figures

1	High level block diagram	5
2	LC Sine Wave Oscillators	6
3	Variable Pitch Oscillator	9
4	A Relaxation Oscillator	10
5	Fixed Pitch Oscillator	11
6	Tuning Sensitivity	12
7	Volume Oscillator	14

8	Mixer Circuit	15
9	Frequency To DC 1	17
10	Frequency To DC 2	17
11	Voltage Controlled Attenuator	19
12	Frequency Visualizer with Tuning Column.	20
13	Frequency Visualizer Block Diagram	20
14	Gain Stage	21
15	Band-Pass Filter Stage	21
16	Band-pass filter Bode Plot	22
17	Amplitude Detection Stage	22
18	LED Visualization Stage	24
19	Frequency Visualizer PCB	25
20	Completed Theremin	26
21	Antennas	27
22	Assembled Circuit Boards	28
23	Assembled Visualizer PCB	29

1 Motivation

The theremin is a great project because it is an uncommon instrument that is fun to watch and play. The theremin is also a completely analog device and gave our team some great experience with oscillators, amplifiers, and signal processing. Along with this general knowledge some of the specific operating principals used in the theremin have applications in many different products. For instance, the capacitive sensing techniques used to control the theremin are seen in many devices such as cell phone touch screens and the heterodyne process used to get the pitch frequency is used in many different signal transmitting and receiving systems.

2 System Architecture

2.1 High Level

The theremin has three main parts which were each designed independently by one member of our team. A block diagram of these parts can be seen in figure 1.

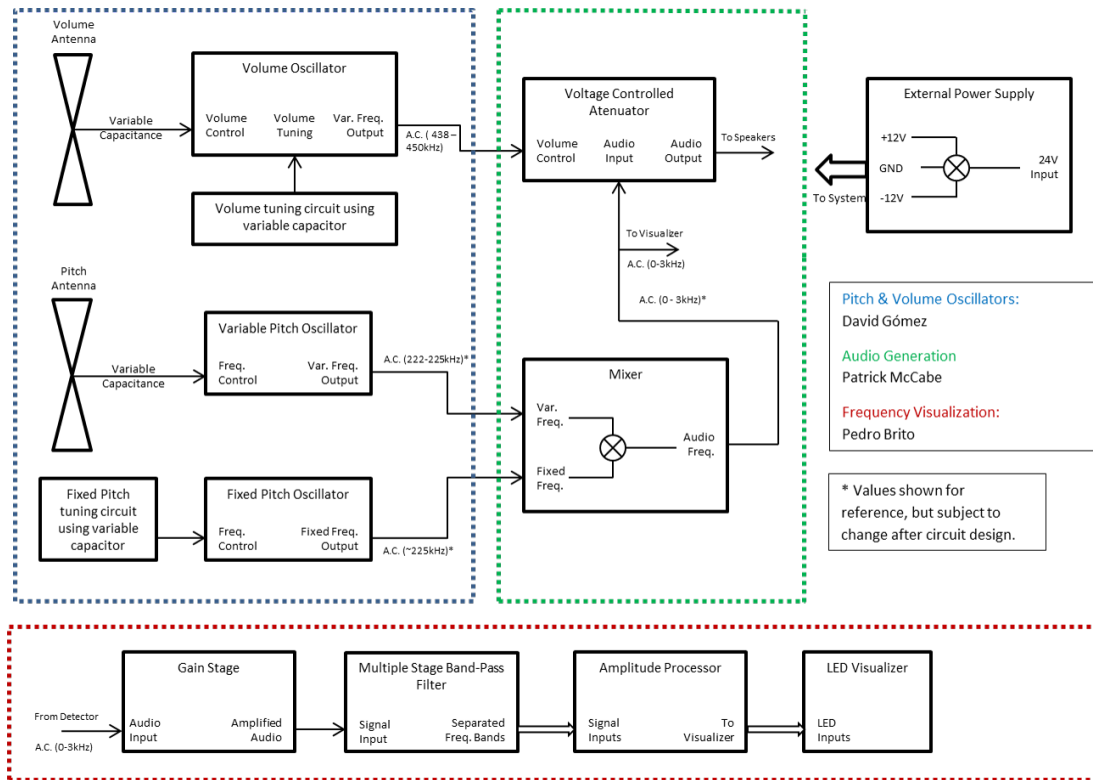


Figure 1: High level block diagram

- **Pitch and Volume Oscillators** A set of oscillators that vary in frequency when the capacitance of the antennas change
- **Audio Generation** Transforms the changing frequency of the oscillators into changes in the pitch and volume of the audio signal

- **Frequency Visualization** Lights up LEDs according to the frequency of the audio signal being produced

With these three parts our team was able to achieve the goal of producing an audio signal at line level that could vary in pitch from $\sim 100\text{Hz}$ - 3kHz and vary in volume from loud to silent. We are also able to visualize a large range of musical notes throughout this pitch range.

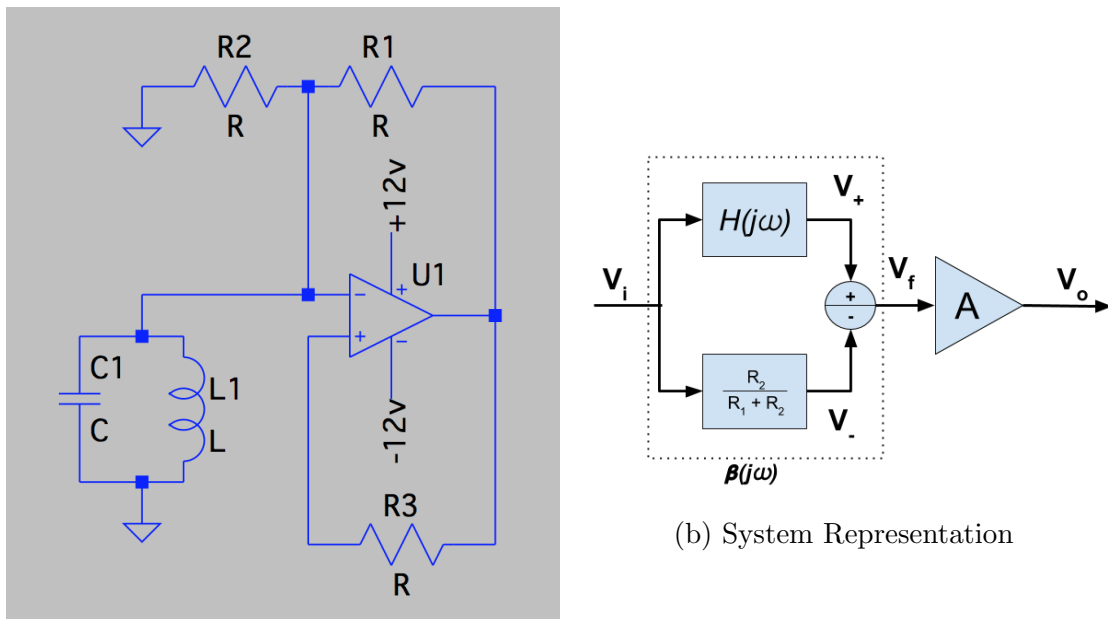
2.2 Pitch Oscillators (David)

2.2.1 Purpose

The pitch oscillator circuits are responsible for detecting the change in capacitance from the pitch antenna (and therefore the change in hand position) and translating it into a changing frequency to be fed into the audio generation circuits to produce a tone.

2.2.2 Theory - Sine Wave Oscillator

A simple way which most theremins use to detect changing capacitance is connecting the antenna to an LC resonant tank oscillator. When the capacitance of the LC tank changes, the resonant frequency of the tank also changes producing the capacitance dependent frequency needed. The sine wave produced by such oscillators is also part of what gives theremins their distinctive sound.



(a) Circuit Representation

(b) System Representation

Figure 2: LC Sine Wave Oscillators

LC tank resonant tanks do not oscillate on their own though. In order to get the tank to oscillate feedback must be applied. In figure 2, two representations of an LC sine wave oscillator circuit can be seen, a circuit and system view. The

system view shown in figure 2b is particularly useful because it ensures that the circuit will oscillate by satisfying the Barkhausen Stability Criterion which states that a system will oscillate under two conditions:

- The open loop gain is unity in magnitude, $|\beta A| = 1$
- The phase shift is zero or a multiple of 2π , $\angle\beta A = 2\pi n, n \in 0, 1, 2, \dots$

The open loop gain of this system is:

$$\frac{V_0}{V_i} = \frac{V_f}{V_i} \frac{V_0}{V_f} = \beta(j\omega)A$$

So to satisfy the gain criterion of Barkhausen:

$$|\beta(j\omega)| = \frac{1}{A} \quad (1)$$

Where A is the open loop gain of the Opamp and $\beta(j\omega)$ is defined as the combined transfer function of the rest of the circuit:

$$\begin{aligned} \beta(j\omega) &= H(j\omega) - \frac{R_2}{R_1 + R_2} = \frac{Lj\omega}{LC(j\omega)^2 R_3 + Lj\omega + R_3} - \frac{R_2}{R_1 + R_2} \\ |\beta(j\omega)| &= \frac{\sqrt{R_2^2 R_3^2 (LC\omega^2 - 1)^2 + R_1^2 L^2 \omega^2}}{\sqrt{(R_1 + R_2)^2 R_3^2 (1 - 2LC\omega^2 + L^2 C^2 \omega^4) + (R_1 + R_2)^2 L^2 \omega^2}} \\ \angle\beta(j\omega) &= \tan^{-1}\left(\frac{R_3}{L\omega - LC\omega^2}\right) \end{aligned}$$

When the circuit is operating at its resonant frequency, the equations simplify and make it clear which values could lead to Barkhausen being satisfied:

$$\begin{aligned} |\beta(j\omega)| &= \frac{R_1 L \omega_0}{(R_1 + R_2) L \omega_0} = \frac{R_1}{R_1 + R_2} \\ \angle\beta(j\omega) &= \tan^{-1}\left(\frac{R_3}{\sqrt{\frac{L}{C}} - 1}\right) \end{aligned}$$

To satisfy the gain criterion of Barkhausen, the ratio of R_1 and R_2 must simply be set to satisfy equation 1. Since A is not known exactly, a potentiometer can be used to adjust the ratio of R_1 and R_2 . To satisfy the phase criterion R_3 must theoretically be set to zero; however, this does not consider the non-ideal components of the circuit and that the opamp may introduce some phase in its gain A that is not shown in the math. Because of this complexity, the simplest thing to do is to make R_3 a variable resistor.

2.2.3 Theory - Detecting Capacitance Change

Section 2.2.2, has shown how make an LC resonant tank oscillate; however, LC values and through that, the resonant frequency must still be selected. This is a challenging problem because a hand moving around the antenna was only able to

create a change in capacitance of around 3pF which must be turned into a frequency change of 3kHz to meet the pitch change requirements in our goals. As will be shown in this section, meeting these requirements requires a careful choice of values.

To begin, the resonant frequency of the oscillator made in section 2.2.2 is given by:

$$F_0 = \frac{1}{2\pi\sqrt{LC}} \quad (2)$$

Rearranging this and taking the partial derivative of frequency with respect to capacitance:

$$\frac{\partial F}{\partial C} = \frac{-1}{4\pi\sqrt{LC^3}} = -\sqrt{\frac{L}{2C}}F_0 \quad (3)$$

These equations make some important design decisions apparent. The derivative shown in figure 3 is negative so frequency decreases as capacitance increases. Since capacitance increases as the hand moves closer to the antenna, one might think the resonant frequency of the LC tank could be set such that it is 3kHz when no hands are near and then the additional 3pF from a hand would be enough to get the necessary frequency drop. This is unfortunately not feasible. The antenna's normal capacitance is around 12pF, to get a resonant frequency of 3kHz an impractically huge inductor of a few hundred henry would be needed. These values would also result in a $\frac{\partial F}{\partial C}$ of only tens of hertz per picofarad, not nearly enough to met the goals set.

From equation 3, it can be can see that our sensitivity to capacitance changes can be increased by increasing our resonant frequency or maximizing inductance for a given resonant frequency. A resonant frequency of 264kHz with a 3.3mH inductor and 110pF capacitor was chosen to give a theoretical $\frac{\partial F}{\partial C}$ of $1.2\frac{kHz}{pF}$ and a range of 3.6kHz.

Of course, this strategy creates a new problem which is that the resonant frequency is now out of the range of human hearing so it cannot be used directly as the audio signal. This problem is solved by adding a second oscillator and heterodyning them. This is discussed further in section 2.4.2; however, at a high level, by adding a second oscillator at a fixed frequency the signals can be mixed to create a new signal containing the difference between their frequencies. This is very useful since another fixed oscillator that runs at the resonant frequency of the variable oscillator connected to the antenna when no hands are near it can be added. When a hand moves closer to the antenna, the variable oscillator's frequency decreases but the fixed oscillator holds its frequency resulting in an increasing frequency difference between them. The frequency difference theoretically will vary from 0Hz to the maximum the variable oscillator's frequency can decrease, in this case 3.6kHz, making it perfect to be directly used as an audio signal.

2.2.4 Circuit - Variable Pitch Oscillator

Figure 3 shows the final schematic of the Variable Pitch Oscillator (VPO). This oscillator uses the topology discussed in section 2.2.2. As stated in section 2.2.3, the the values used in the resonant tank (L_2 and C_5) are 3.3mH and 110pF for a theoretical resonant frequency of 264kHz. The antenna is also shown connected through L_1 . Since the antenna acts as a capacitor this gives us a series LC tank connected to our parallel tank which is a commonly used technique in theremin design to couple the antenna capacitance to the resonant tank. Unfortunately, it was not simple to calculate the resonant frequency of the total system and L_1 had to be determined experimentally but was initially sized to give a 264kHz resonant frequency with the 12pF antenna capacitance to match the LC tank. Upon connecting the tank LC to the antenna LC and soldering the circuit onto a proto-board, a resonant frequency of about 225kHz was found. This was likely caused by an inability to properly model the antenna LC and also parasitic capacitance from connecting wires, which are likely on the order of pico farads which is a problem when the tank capacitance and antenna capacitance are 110pF and 12pF respectively. While the resonant frequency fell short of what what predicted, the range was within our expectations. In the playing area, a 3kHz frequency change was easily produced. The total frequency change was even more, but involved placing a hand very close to the antenna. The area directly near the antenna is unusable for playing since the capacitance change is not nearly as linear as it is farther from the antenna.

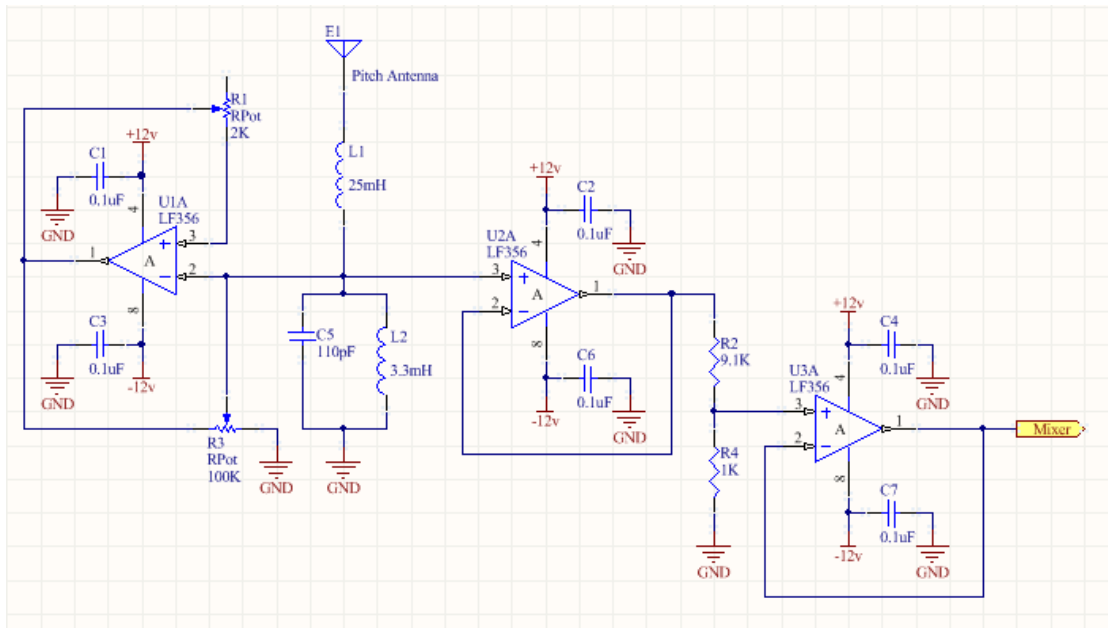


Figure 3: Variable Pitch Oscillator

The sine wave output is taken directly from the LC tank and buffered through U2A before going through a voltage divider and buffer that drops the $24V_{p-p}$ signal down to a $2.6V_{p-p}$ signal needed for the mixer input. LF356 opamps are used exclusively because of their high speed and availability in the 6.101 lab.

There are also $0.1\mu F$ capacitors placed closely to the power pins of all the op amps to help provide a clean power supply.

2.2.5 Theory - Fixed Frequency Oscillator

As discussed in section 2.2.3, heterodyning a high frequency fixed and variable oscillator allows us to sense the very small antenna capacitance change and transform it into an audio signal. The design of this oscillator has three important considerations.

1. Must be highly stable - Since the audio frequency is the between the fixed and variable oscillators, frequency drifts of even a few hundred hertz could be noticeable. Since this oscillator operates in the 200kHz range this necessitates high stability
2. Frequency Tunability - The fixed oscillator should be set to the resonant frequency of the variable oscillator; however, because the ambient capacitance and frequency drift of the oscillators can change that resonant frequency, the fixed pitch oscillator must be tunable
3. Sine or Square wave output - The mixer accepts both of these waveforms as shown in section 2.4.2

An oscillator topology that satisfies all three of these constraints is the relaxation oscillator, shown in figure 4. This oscillator uses a comparator and a RC circuit to generate a fixed frequency square wave that switches between the positive and negative voltage rails. When the output voltage of the comparator is at the positive supply rail, C_1 is charged through R_3 producing an increasing voltage on the inverting input of the comparator while the non-inverting input is held at a constant voltage set by the R_1 and R_2 voltage divider. After enough time, C_1 will have charged to a greater voltage than the constant voltage on the non-inverting input at which point the comparator's output will go to the negative supply rail. After this, C_1 will begin to discharge through R_3 until it's voltage reaches the voltage at the non-inverting input of the comparator causing the output to go high again at which point the whole cycle will repeat forever.

Since the comparator flips between high and low, a square wave output with a frequency given by equation 4 is produced.

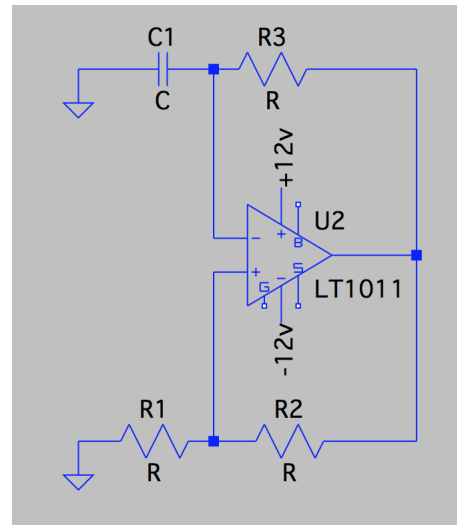


Figure 4: A Relaxation Oscillator

$$f = \frac{1}{2\ln(3)R_3C_1} \quad (4)$$

From equation 4 it is also evident that the frequency is only dependent on the R_3C_1 time constant which is highly convenient because frequency tunability can be added by simply replacing R_3 with a potentiometer. Since it does not affect the frequency, we can also simply set R_1 equal to R_2 .

2.2.6 Circuit - Fixed Pitch Oscillator

Figure 5 shows the complete schematic of the Fixed Pitch Oscillator (FPO). The first thing to note is the choice of the U1A as an LT1011A comparator for use in the relaxation oscillator designed as in section 2.2.5. This was the fastest comparator available in lab. The speed of the comparator actually ended up being an issue since at $\sim 225\text{kHz}$ (the frequency required to match the VPO) the sharp transition between the $\pm 12\text{V}$ power rails was very difficult to generate without seeing rounded edges on the square wave. Even the LT1011A was not fast enough to generate good square waves at 225kHz but fortunately since the mixer only required a $2V_{p-p}$ square wave the rounded square wave was able to be clipped which left a very nice square waveform. The clipping circuitry was composed of first a voltage divider with R_{11} and R_{14} which was needed because LF356 U2A had issues accepting the full signal. After this divider, there is a buffer and then a set of diodes. These diodes clip the output signal at 2V and -2V because the V_f of the 1N914 is 1V and there are two in series in opposite polarities relative to ground.

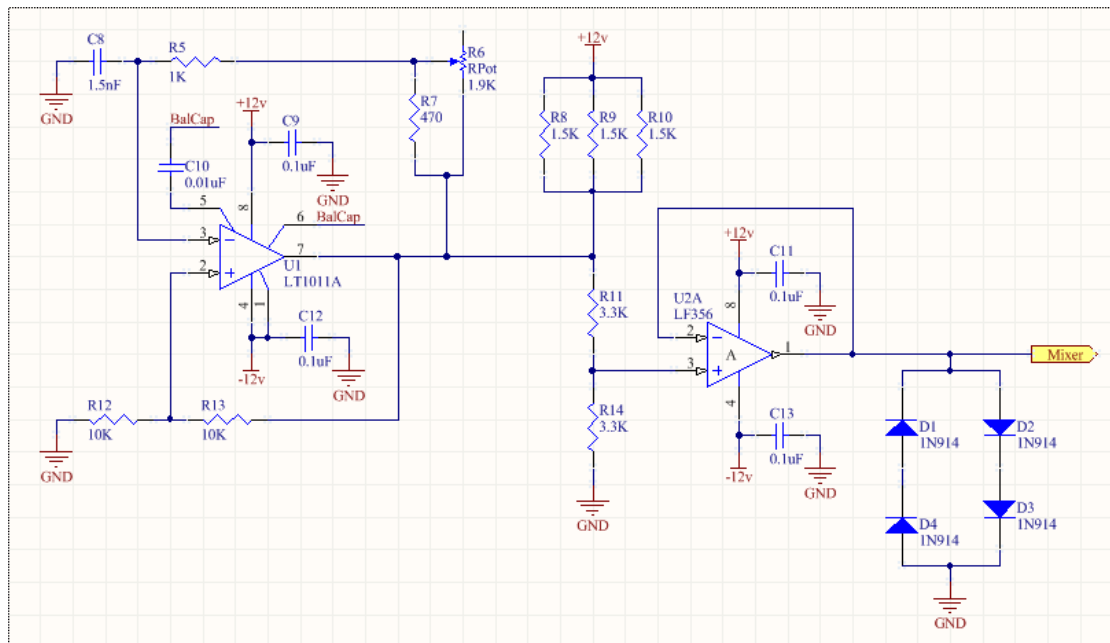


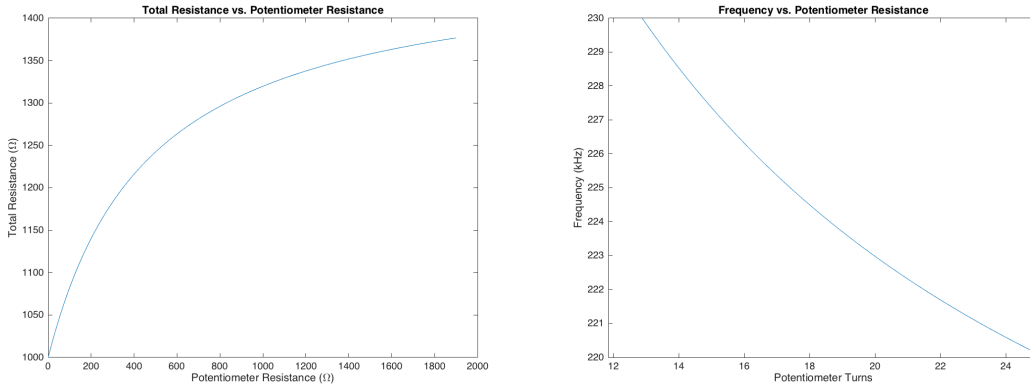
Figure 5: Fixed Pitch Oscillator

The LT1011A has an open collector output so connecting pin 1 to -12V and adding a pullup resistor to +12V allows us to get rail to rail output. The pullup

resistor is actually three pullup resistors, R_8 , R_9 , and R_{10} , in parallel for a combined 500Ω . All three of these resistors were $\frac{1}{2}$ Watt and placing them in parallel reduced the current seen by each one. This was done after finding that a single 510Ω $\frac{1}{8}$ Watt resistor got very hot and since heat could be a cause of frequency instability we wanted to avoid this.

C10 was recommended to be placed between the balance pins by the data sheet. Other capacitors by the power rails are for supply filtering.

The frequency of this oscillator was set to be around 225kHz which as seen in section 2.2.4 is the frequency that could be realized for the VPO. In figure 5, C_8 and the resistor combination made by R_5 , R_6 , and R_7 replace C_1 and R_3 from equation 4 to determine the frequency of oscillation. C_8 is set to be the low value of 1.5nF as a low capacitance was found to help improve the quality of the square wave. The purpose of the resistor combination is to maximize the tuning sensitivity of the oscillator. It was shown in 2.2.5 that R_3 from equation 4 could be replaced with a potentiometer to provide tunability. The challenge with tuning is that it must be highly sensitive because even slight variations of a few hundred hertz create large change in the pitch. Early testing with a half turn potentiometer was unsuccessful because it was difficult to turn the potentiometer a small enough amount to get the desired frequency change. To get better resolution, a 25 turn potentiometer was used instead so that the same amount of turn produced less of a change in resistance and therefore less of a change in frequency. This resolution can be improved even further by placing a resistor in parallel with the potentiometer and taking advantage of the non-linearity of the total resistance of resistors in parallel.



(a) Change in total resistance through po- (b) Change in frequency per turn of poten-
tentiometers full range tiometer

Figure 6: Tuning Sensitivity

Figure 6a shows the change in the total resistance of the resistor combination as given by the following equation:

$$R_{total} = R_5 + \frac{R_6 R_7}{R_6 + R_7}$$

When the potentiometer R_6 is near its upper limit the change in total resistance is small compared to the change in potentiometer resistance. By properly sizing

R_5 and R_7 , the frequency of 225kHz can be centered in this low slope area to provide maximum tuning resolution. As shown in figure 6b, near 225kHz a tuning resolution of around 1kHz per turn is achieved.

The only remaining components are R_{12} and R_{13} which were made identical because as discussed in section 2.2.5 their value does not affect the frequency.

2.3 Volume Oscillator (David)

2.3.1 Purpose

Similar to the pitch oscillators, the volume oscillator (VO) is responsible for detecting a capacitance change to produce a change in the audio output, in this case the volume. However, instead of producing a variable frequency audio signal, the VO only needs to produce a frequency changing sine wave which as later seen in section 2.4.4 will be transformed into a changing DC value to attenuate the volume of the audio signal.

2.3.2 Design - Volume Oscillator

Since the VO only needs to produce a sine wave that changes frequency depending on the capacitance of the antenna and hand, the oscillator topology created in 2.2.2 can be used with only some minor modifications. No heterodyning is required because the voltage attenuation circuitry is capable of dealing with the signals on the order of 100kHz range necessary to give good sensitivity to changing capacitance. A resonant frequency of 455kHz with a 1mH inductor and 120pF capacitor in parallel with a 2-100pF variable capacitor was selected. The higher frequency was chosen to give a higher theoretical $\frac{\partial F}{\partial C}$ of $1.87 \frac{\text{kHz}}{\text{pF}}$. The variable capacitor was used because unlike in the pitch oscillators, where tuning to the ambient capacitance could be done through the FPO, the tuning for the VO must be done with the oscillator itself.

2.3.3 Circuit - Volume Oscillator

Figure 7 shows the complete schematic of the VO. The design is almost identical to the VPO from section 2.2.4. L_1 and L_2 were also chosen experimentally as they were with the VPO. The VO was able to achieve a resonant frequency of 450kHz and had a range of almost 12kHz.

The output of the LC tank is buffered and then scaled down just as with the VPO; however, instead of a fixed voltage division we use potentiometer R_2 to allow the output scaling to be modified.

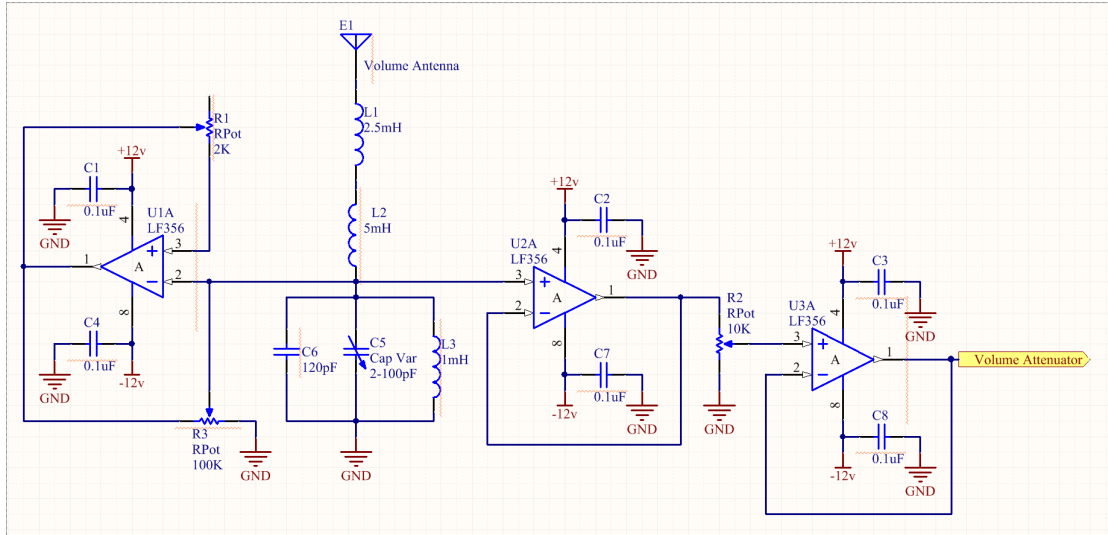


Figure 7: Volume Oscillator

2.4 Audio Generation (Patrick)

2.4.1 Purpose

The audio generation circuitry has two roles: creating the audible pitch and controlling its volume. The pitch has to be extracted from the difference in frequency of the fixed pitch oscillator and the variable pitch oscillator. The volume has to be controlled using the variable frequency output of the volume oscillator.

The audio generation circuitry can be broken into three modules: a signal mixer, a frequency to DC voltage converter, and a voltage controlled attenuator.

2.4.2 Theory - Signal Mixing

The fixed pitch oscillator (FPO) is tuned such that its output frequency is equal to the output of the variable pitch oscillator (VPO) when there is not a hand near the pitch antenna. This means that the difference in frequency between these two signals is $\sim 0\text{Hz}$ when the theremin is not being played. As the musician moves their hand closer to the pitch antenna the variable pitch oscillator's output will increase in frequency. This causes the difference between the VPO's frequency and the FPO's frequency to increase. This difference reaches approximately 3KHz at its maximum. By building circuitry that can generate a sinusoid at this frequency difference the theremin can generate a pitch with range $\sim 0\text{Hz}-3\text{KHz}$. Performing this operation is the job of the mixer circuit.

Given two sinusoids, y_1 and y_2 , with two different frequencies, f_1 and f_2 , the multiplication of $y_1 * y_2$ will lead to the creation of two sinusoids. One of these sinusoids will have frequency $f_1 - f_2$ and the other will have frequency $f_1 + f_2$. This is the principle of mixing two signals to generate a new signal of a different frequency. This multiplication operation is what the mixing circuitry uses to generate a sinusoid with a frequency equal to the difference between the VPO's frequency, f_{VPO} , and the FPO's frequency, f_{FPO} . Along with outputting

a sinusoid of frequency $f_{VPO} - f_{FPO}$, which ranges from 0Hz-3KHz, the multiplication also generates a signal with a frequency of $f_{VPO} + f_{FPO}$, which is not the frequency we are interested in. However, this sum frequency is approximately 500KHz, which means it can be easily filtered out since it is significantly higher than the audible range.

It is worth noting that it is not necessary for both of the input signals of the mixer to be sinusoids. In fact, our design uses a sinusoid for the variable pitch oscillator and a square wave for the fixed pitch oscillator. The generated output of the mixer is still a sinusoid with a frequency equal to the difference of the two inputs. The reason this works can be seen in the fourier transform of a square wave:

$$square(t) = \frac{4}{\pi} \left(\sin(\omega_0 t) + \frac{1}{3} \sin(3\omega_0 t) + \frac{1}{5} \sin(5\omega_0 t) + \dots \right) \quad (5)$$

Due to the low-pass filtering on the mixer's output only the sinusoid at the square wave's fundamental frequency is of importance.

2.4.3 Circuit - Signal Mixing

The signal mixing circuit can be seen in figure 8.

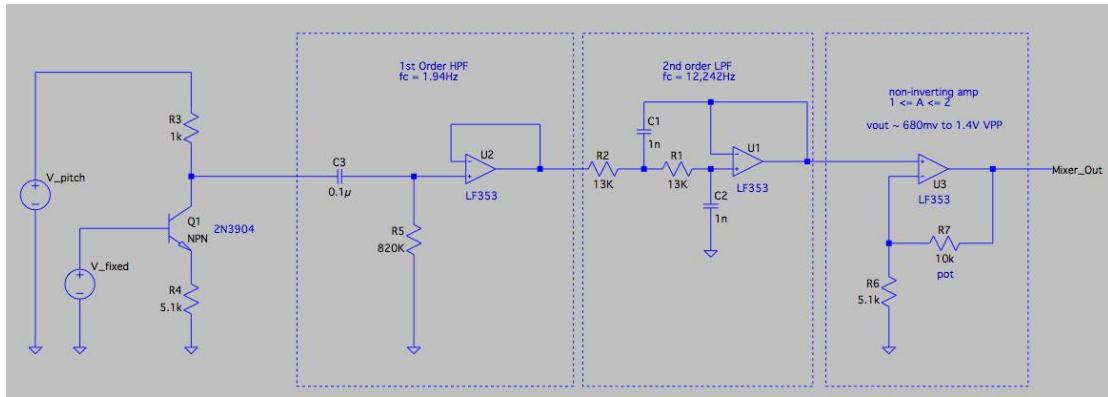


Figure 8: Mixer Circuit

This is a single transistor mixer topology. The mixer circuit gates the output of the variable pitch oscillator by using a 2N3904 transistor and the output of the fixed pitch oscillator as the control signal. When the fixed pitch oscillator's output is a square wave, as it is in our case, this has the effect of producing an output that is the multiplication of the VPO's output by the FPO's output.

The output of this multiplication is fed into a first order low-pass RC filter. This filter is used to remove any DC bias from the multiplication output. The -3dB point of this filter is ~ 2 Hz. This was selected to be a decade below the lowest audible frequency, which is 20Hz. The output of this filter is buffered before going into the next stage of the mixer.

The next stage of the mixer removes the high frequency sinusoids that are produced in the multiplication operation. This is done using a Sallen-Key second order low-pass filter. The cut-off frequency of this filter is 12,242Hz. This cut-off

frequency is not quite a decade above our highest producable frequency (3KHz). This means the design sacrifices some attenuation of the higher frequencies for a reduction in unwanted signals.

Lastly, there is a variable, non-inverting voltage amplifier stage on the output of the mixer. The gain of this stage can be adjusted using a 10K potentiometer. The voltage gain can range from 1 to 2. This stage sets the maximum amplitude of the final line-level audio signal - thereby also setting the maximum volume that can be generated. The magnitude of the resulting mixer output signal can be adjusted from $680mV_{PP}$ to $1.3V_{PP}$. The consumer grade line-level standard for audio uses a maximum voltage of 890mVPP. So this adjustable gain stage allows the instrument to be tuned to output a line-level standard voltage. While this stage outputs a maximum volume signal, the voltage controlled attenuator (the details of which are covered in a later section) allows the musician to decrease the volume when playing the instrument.

2.4.4 Theory - Frequency to DC Voltage Conversion

The musician can control the volume of the instrument's audio signal by moving their hand near the volume antenna. The volume oscillator outputs a sinusoid which varies in frequency based off of the distance of the musician's hand to the volume antenna. This varying frequency signal needs to be converted into a DC voltage so that it can be used to control the voltage controlled attenuator. It is the voltage controlled attenuator that will adjust the volume of the audio.

The method used for converting a variable frequency signal to a variable DC voltage signal can be broken down into two steps. The first step is to pass the variable frequency signal through a low pass filter, with the goal of attenuating the signal. The second step is to pass the output of the filter through a peak detector circuit.

The first step of this process uses a low pass filter with a cut-off frequency below that of the input signal's frequency. The output of this filter will be an attenuated version of the input signal. The magnitude of the attenuation will depend on the input frequency. The higher the input frequency the more the signal will be attenuated. So as the player moves their hand near the volume antenna the frequency of the volume oscillator changes and the magnitude of the filter's output signal changes.

The next step of this process passes the output of the low-pass filter through a peak detector. The peak detector detects the magnitude of the attenuated signal and produces a corresponding DC voltage. So after this step the variable frequency signal has been converted to a variable DC voltage signal.

2.4.5 Circuit - Frequency to DC Voltage Conversion

The frequency to DC voltage converter circuit can be seen in figure 9 and 10. For visibility this circuit has been cut in two. The output of figure 9 flows into the input of figure 10.

The first block in figure 9 is a third order Sallen-Key low-pass filter with a cut-off frequency of 300KHz. This filter implements the low-pass filter stage

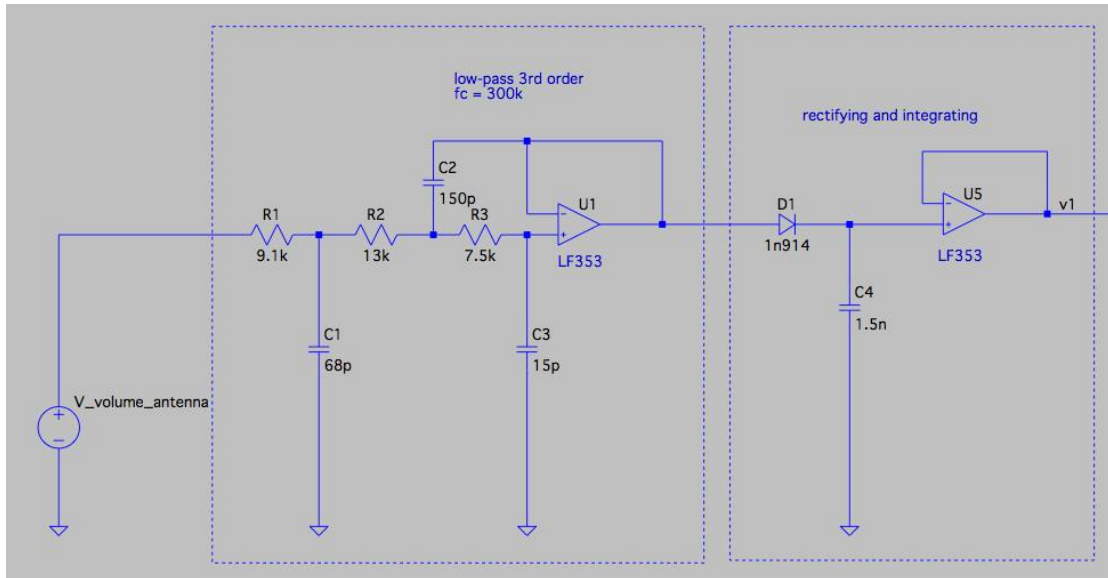


Figure 9: Frequency To DC 1

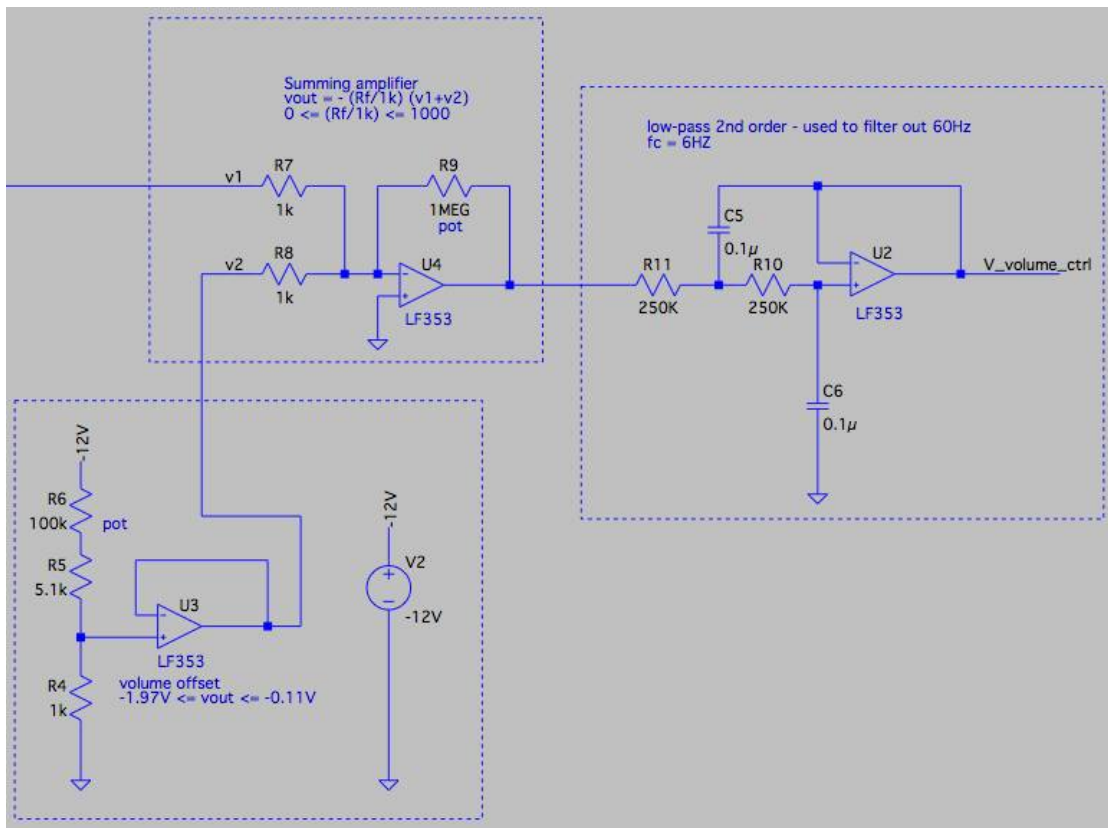


Figure 10: Frequency To DC 2

discussed in section 2.4.4. The third order filter provides a -60dB/dec attenuation above the cut-off frequency. The cut-off frequency was selected to be lower than the 450KHz volume oscillator signal, but not low enough to vastly attenuate the signal. This filter was chosen to be a third order filter to produce a large range of attenuation over the relatively small frequency change produced

by the volume oscillator.

The second block in figure 9 is a peak detector circuit. This circuit converts the changing frequency/amplitude signal from the output of the low-pass filter into a DC voltage. It does this by first rectifying the signal using a 1N914 diode and then low-passing that rectified signal. The output of this detector circuit is buffered before being passed to the next block in the circuit. The next block in the circuit can be seen in figure 10. This block, at the top left of the figure, is an adder circuit. The transfer function for this block is:

$$v_{out} = -\frac{R_9}{1000}(v_1 + v_2) \quad (6)$$

R_9 is a 1Meg potentiometer. So this stage has a voltage gain that ranges from 0 to 1000. This adder allows the instrument to be tuned such that the output volume is silenced when there is no hand near the volume antenna. When there is no hand near the volume antenna there is a non-zero DC voltage produced at the output of the peak detector. This signal needs to be shifted down to 0V so that it will produce silent audio when used to control the voltage controlled attenuator.

So the output of the peak detector, v_1 , is shifted down by the negative voltage v_2 . The voltage v_2 is generated by the circuit block at the bottom of figure 10. This block is a voltage follower that is following the output voltage of an adjustable voltage divider. This adjustable voltage divider uses a 100K potentiometer to adjust its output between -0.11V and -1.97V. This range was experimentally selected.

As a part of the theremin's tuning process, the gain of the adder is tuned such that it outputs a voltage of -10V when the player's hand is very close to the volume antenna. This control voltage corresponds to no attenuation at the voltage controlled attenuator and full volume at the audio output.

The last block of this stage is the circuit block on the right of figure 10. This is a second order low-pass Sallen-Key filter meant to filter out 60Hz noise. The cut-off frequency of this filter is set at 6Hz, or a decade below 60Hz. This was added after it was clear that the high gain of the adder stage was also amplifying 60Hz noise. The output of this stage is the DC voltage that is used to control the voltage controlled attenuator.

2.4.6 Theory - Voltage Controlled Attenuator

The voltage controlled attenuator uses the DC control voltage from the frequency to DC converter to adjust the volume of the audio signal. The attenuator provides a theoretical gain between 0 and 1. Since the amplitude of the output from the mixer is set to be at full volume according to the line-level specification, this attenuator is all that is need to provide a full range of volume. This device can be constructed using a voltage divider where the bottom resistor has a variable resistance based on the control voltage.

2.4.7 Circuit - Voltage Controlled Attenuator

The voltage controlled attenuator circuit can be seen in figure 11.

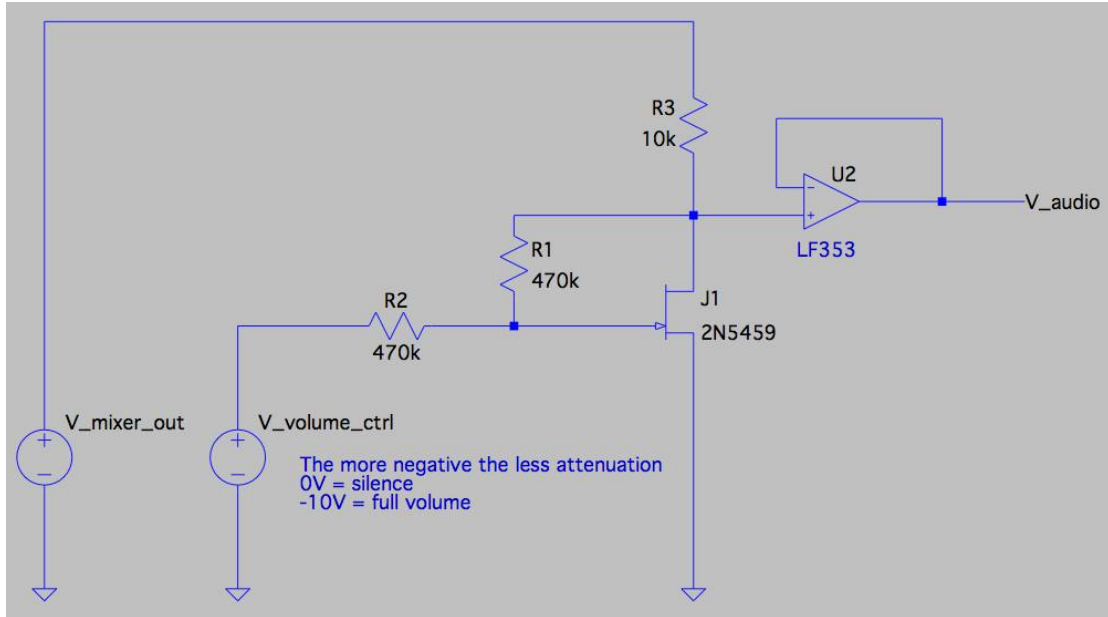


Figure 11: Voltage Controlled Attenuator

The circuit is a voltage divider where the output is of the form:

$$v_{audio} = \frac{R_{var}}{R_{var} + R_3} \cdot v_{mixer\ out} \quad (7)$$

R_{var} is the variable resistance due to the 2N5459 JFET. This JFET is used as a variable resistor whose resistance is controlled by the control voltage produced by the frequency to DC converter. A control voltage of 0V provides the smallest JFET resistance thereby creating the most attenuation and producing a silent audio signal. A control voltage of approximately $-10V$ provides the largest JFET resistance thereby creating no attenuation and producing a full volume audio signal that has the max line-level signal. R_1 and R_2 couple a small amount of the drain signal to the gate of the JFET. This allows the circuit to handle a large input voltage swing and still provide a relatively linear response. Lastly there is a buffer on the output of this stage. The output voltage produced here is the final audio signal used to drive a battery powered speaker.

2.5 Frequency Visualization (Pedro)

2.5.1 Purpose

The Frequency Visualizer serves to give the player a visual representation of where in the frequency spectrum he or she is playing. This is done by having an array of LED columns sensitive to different frequency bands, as shown in figure 12. Each column then has a row of LEDs that lights up incrementally the closer the Theremin output frequency gets to its center frequency. A calibrating row is also added, which responds to a narrow frequency range around the standard tuning frequency A440.

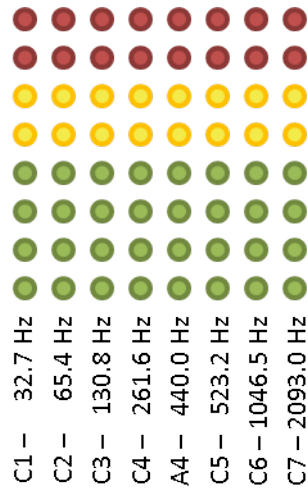


Figure 12: Frequency Visualizer with Tuning Column.

2.5.2 Design

The visualizer has four main stages: a gain stage, a band-pass filter stage, an amplitude detection stage, and a visualization stage. Figure 13 shows the aforementioned stages in a block diagram.

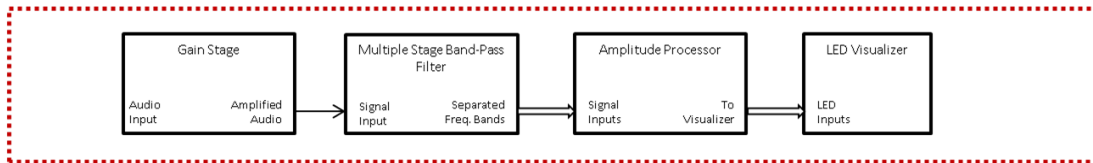


Figure 13: Frequency Visualizer Block Diagram

2.5.3 Circuit - Gain Stage

The gain stage serves to amplify the audio input from line voltage to approximately $22V_{p-p}$ (OpAmp output rail-to-rail voltage). It also gives the added benefit of having a high input impedance into the visualizer circuit. The gain stage was done by implementing a simple non-inverting op-amp circuit, shown in figure 14, with a variable resistor in the feedback path. A variable gain from 8 to 35 is recommended in order to support voltage inputs from 0.6V to 2.5V. The gain of this stage is 1 to $\frac{Pot1}{RG}$.

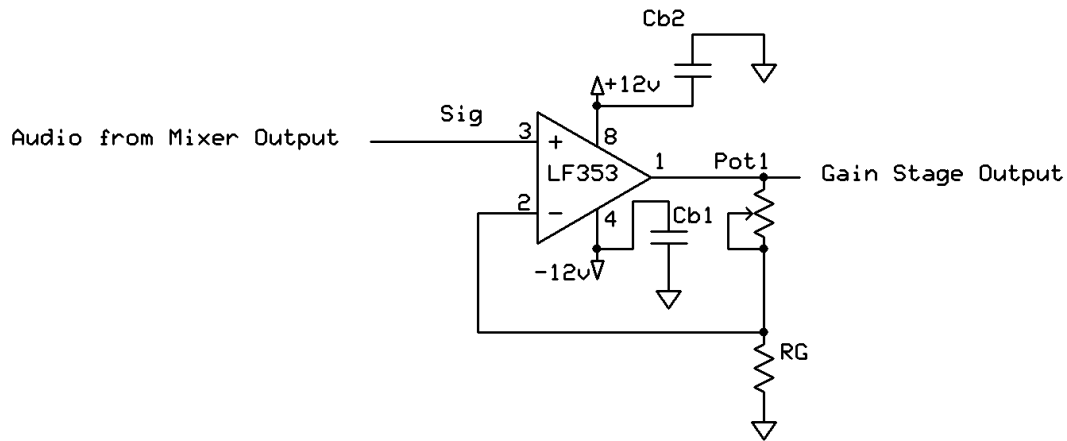


Figure 14: Gain Stage

2.5.4 Circuit - Band-Pass Filtering Stage

In order to be able to visualize different frequency ranges they first have to be detected from the audio input. This is done by having a band-pass filter for each frequency range of interest. A band-pass filter is in essence a combination of low-pass and high-pass filters set up such that only the frequencies of interest are returned, as can be seen in figure 15. In the visualizers case, we want the low-pass and high-pass filters to resonate at the same frequency in order to have a recognizable change in the output of the filter as the frequency changes.

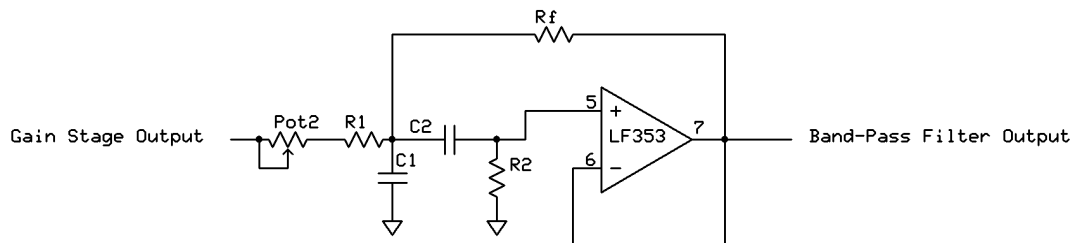


Figure 15: Band-pass filter with feedback and potentiometer for center frequency adjustment.

Figure 16 shows a bode plot of the nine band-pass filters used for the project. Each band-pass filter is centered at the frequency shown in the legend of figure 16. For each band-pass, the output voltage decays as the input deviates from the center frequency.

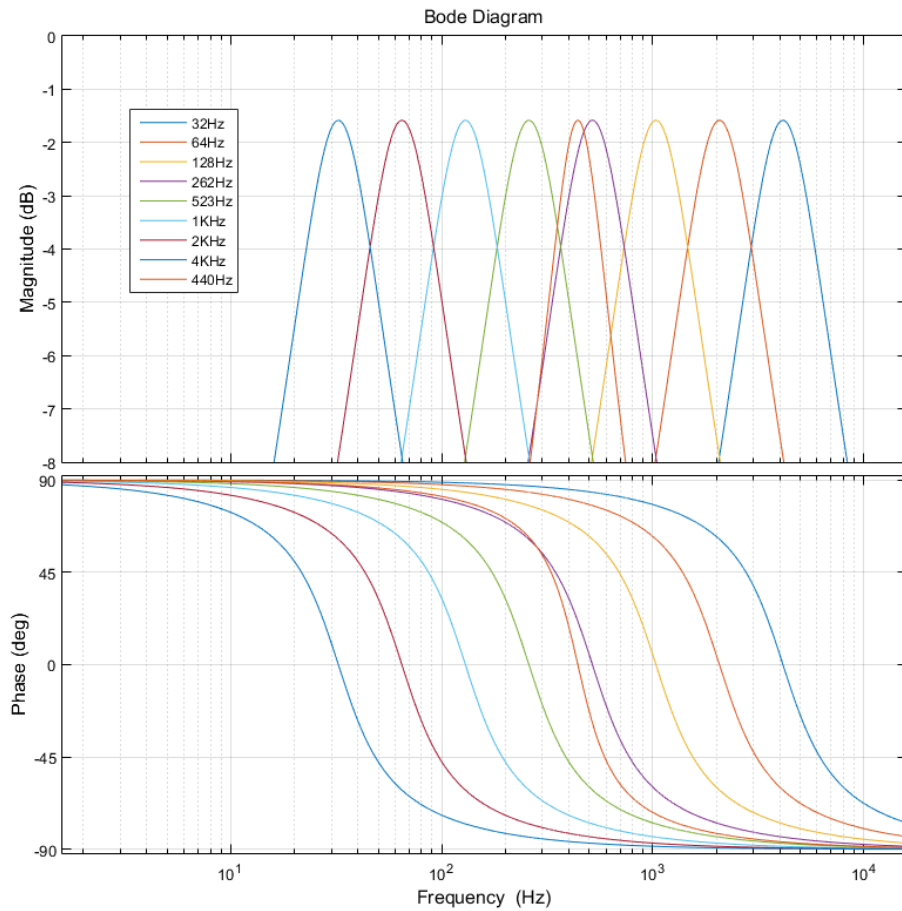


Figure 16: Band-pass filter Bode Plot

2.5.5 Circuit - Amplitude Detection Stage

The amplitude detection stage, shown in figure 17, is in essence a rectifier. As the amplitude of the band-pass filter output decreases, so does the voltage output of the rectifier. Appropriate resistor and capacitor values were chosen such that the rectifier output was sensitive to any changes in the input.

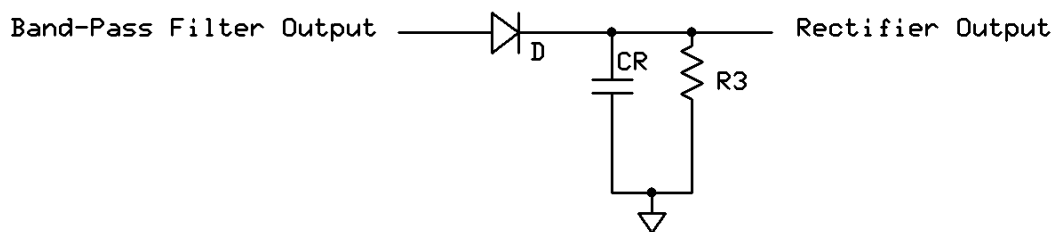


Figure 17: Rectifier diode coupled to a RC circuit for proper detection

2.5.6 Circuit - LED Visualization Stage

The LED visualization stage takes the voltage output of the rectifier (which can range approximately from 0 to 10V) and turns on an array of LEDs incrementally as the voltage goes up. There are two ways of doing this:

- Using astable configured operational amplifiers which turn on respective LEDs after certain thresholds.
- Using a current controlled LED and diode ladder which turns LEDs on incrementally as the current is increased.

The op-amp circuit has the advantage of turning respective LEDs on at precise voltages; however, the need of one op-amp per LED increases cost and valuable space in a PCB. On the other hand, the current controlled ladder shown in figure 18 is extremely compact and simple, with the added drawback of heat dissipation in the current control circuitry and less control of the LED array.

In order to save space in the PCB and simplify the circuit, the current controlled LEDs were deemed to be the best option. It consists of an N-FET driven by the rectifier output. Its source is connected to a variable voltage divider and its drain drives the current controlling PNP transistor. By adjusting Pot3 the amount of current that goes through R4 at a specific rectifier voltage can also be adjusted. This in turn changes the voltage of the PNP gate and how much current goes through it. Pot3 is used when the rectifier (and band-pass filter) output is at its maximum, setting how many LEDs turn on at the rectifiers peak voltage and how sensitive the LEDs are to a rectifier voltage change.

2.5.7 Visualizer Circuit Schematic and PCB

After deciding all of the critical design choices the whole circuit was laid out in a schematic and tested on a breadboard. Testing on the breadboard was successful after the first trial and a final schematic which was composed of all of the previous subsections.

After successful testing it was decided that it would be ideal to construct the whole visualizer on PCBs in order to attain a compact and elegant visualizer. Each visualizer array took a whole breadboard, so nine would have been needed for the whole circuit. With the PCBs, three visualizer arrays could be placed on one board. Three PCBs were ordered for a total of nine arrays.

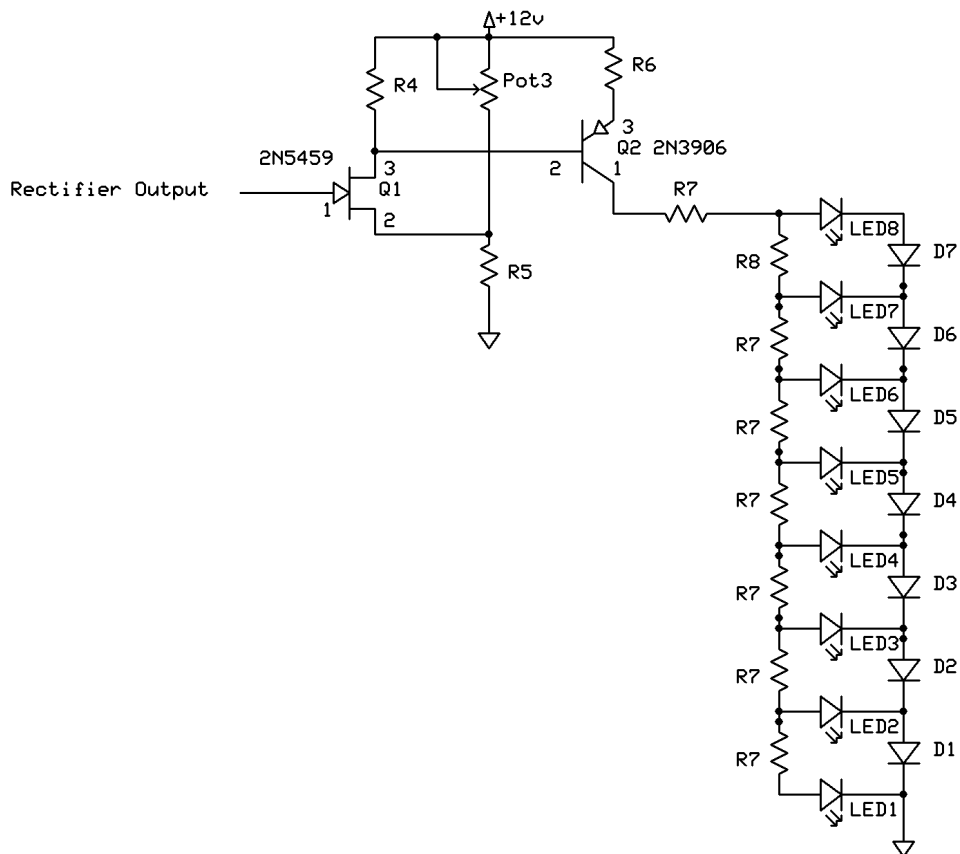


Figure 18: Voltage controlled visualizer circuit with current control driven LED array

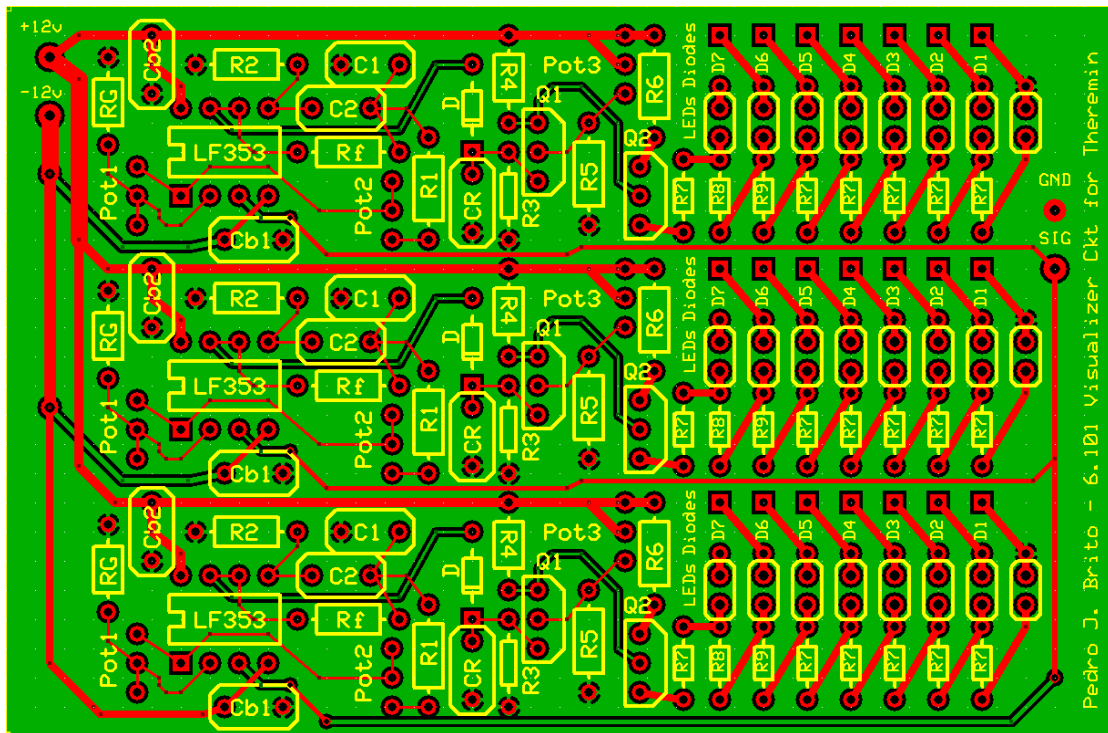


Figure 19: Frequency Visualizer PCB

3 Integration

Once designed and built, the completed subsections of the theremin were all placed together onto one wooden board as shown in figure 20. The frequency visualizer boards were not attached the wooden board and were placed off to the the side.

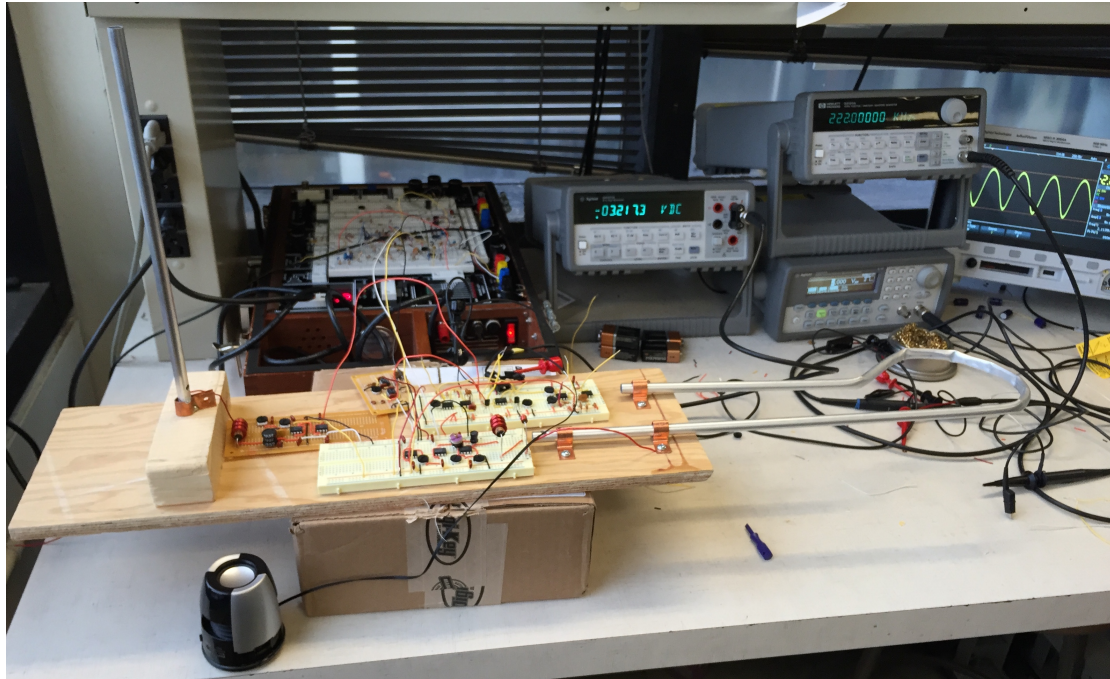
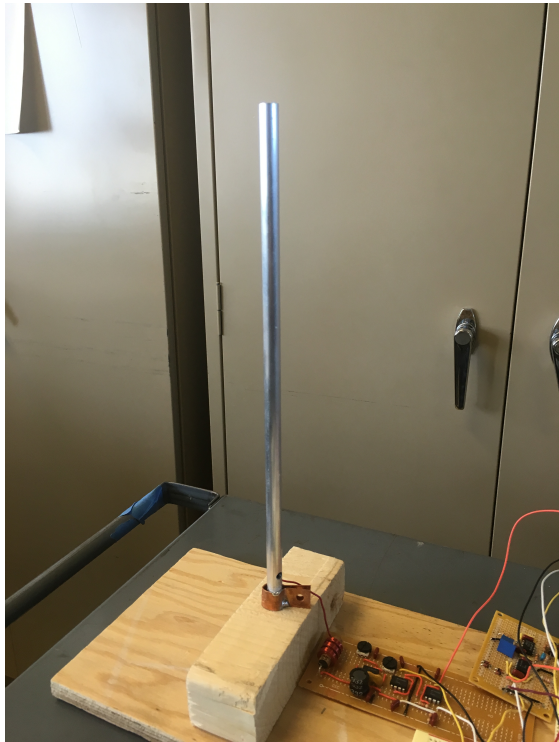


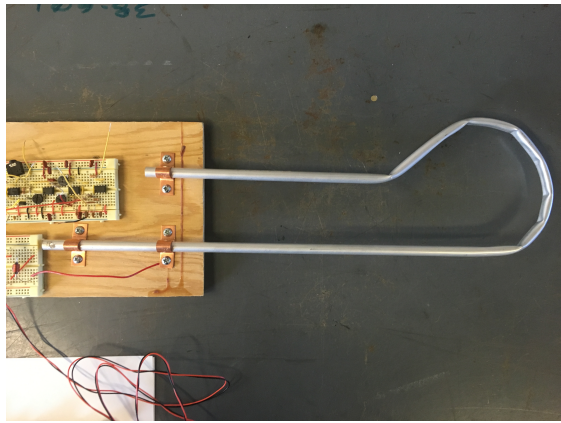
Figure 20: Completed Theremin

The oscillators and audio generation circuitry were powered off of the labkit power supply. The frequency visualizer boards drew more power than the labkit was able to supply so they had to be powered with separate bench supplies.

The antennas shown in figure 21 were also connected to the wood board. The antennas were made out of $\frac{3}{8}$ " aluminum tube. The volume antenna was bent into a U like shape which from research into theremin design is said to increase sensitivity when the hand close. The antennas are perpendicular to each other and spaced widely apart to minimize the interaction the pitch controlling hand might have on the volume and vice versa.



(a) pitch



(b) volume

Figure 21: Antennas

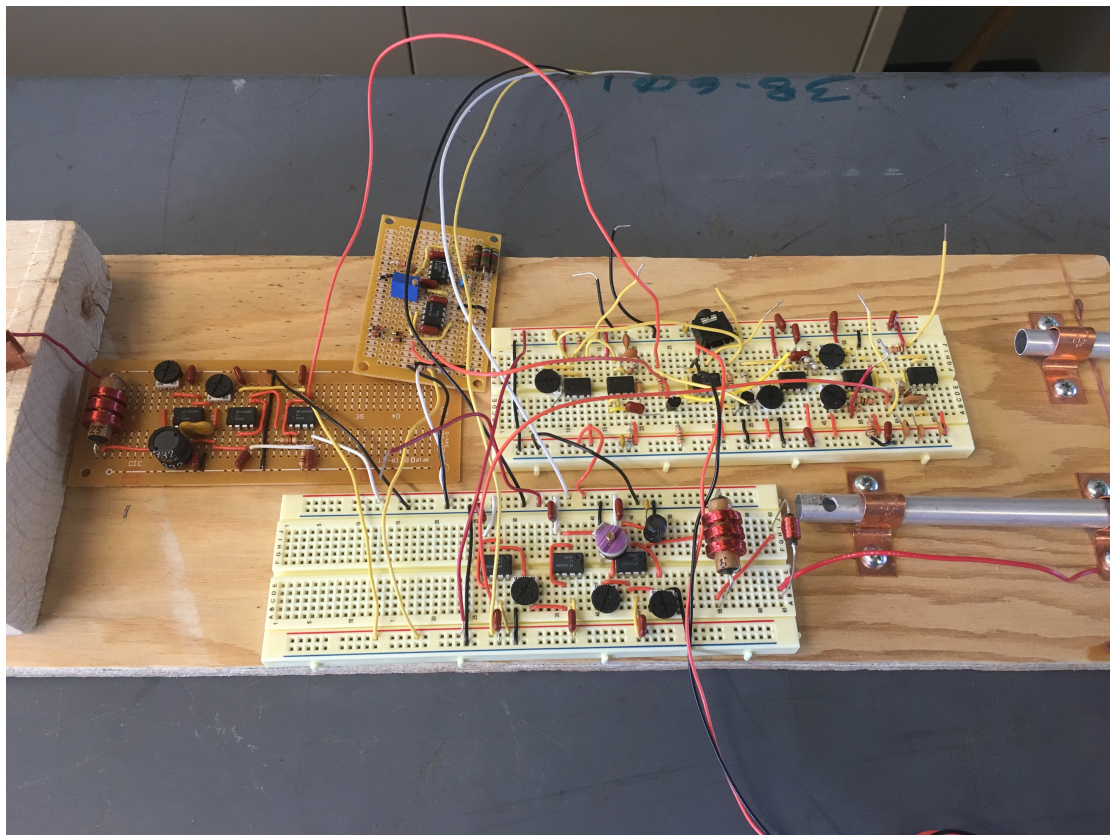


Figure 22: Assembled Circuit Boards - starting at the top going clockwise: FPO, Audio Generation, VO, VPO

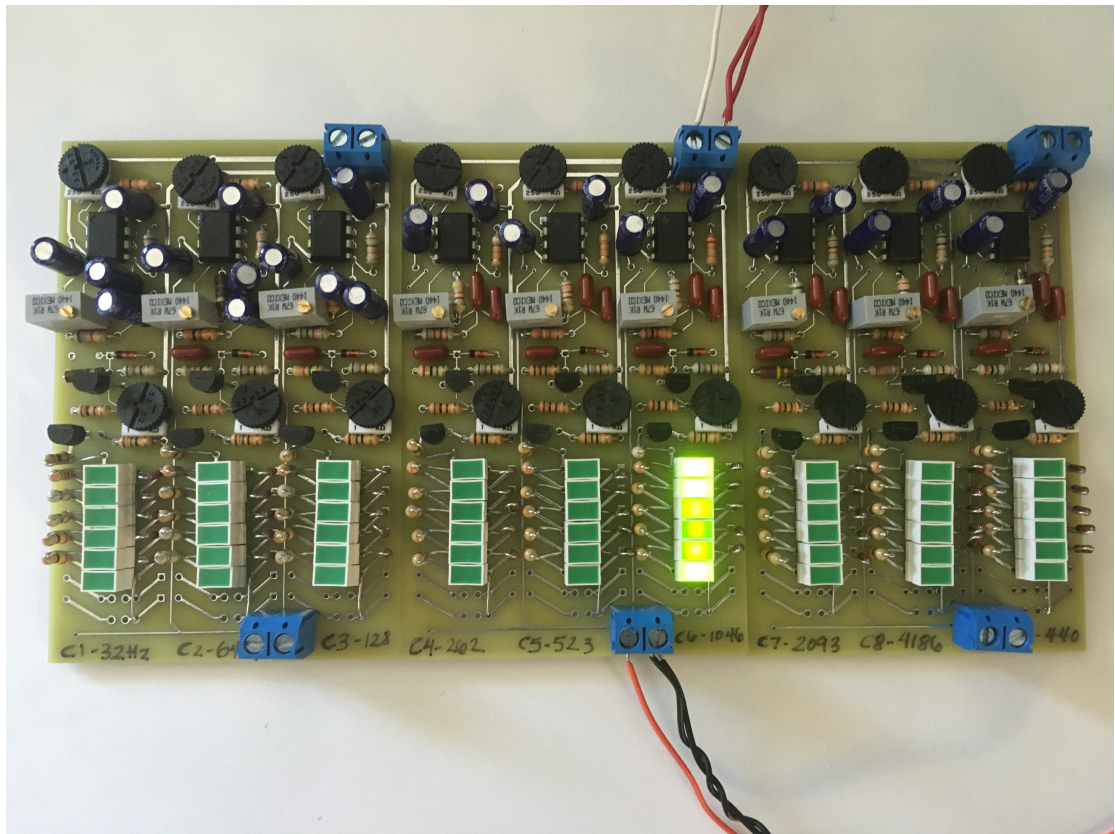


Figure 23: Assembled Visualizer PCB with a column of LEDs illuminated

4 Conclusion

Many theremin schematics are available online; however, since it is such an old instrument, most detailed schematics are greatly outdated. With this project we were able to achieve a modernized version of the theremin which utilizes higher precision components and techniques, such as op-amp based oscillators. Another distinction of our theremin is the addition of visualization. This allows the user to easily tune the theremin and have a clear visualization of what frequency range they are playing in. One of the greatest challenges of this project was adapting to the properties of the constructed antennas. By having accessibility to the antennas from an early phase we were able to design around these properties and achieve our goal. The broad knowledge that was gained throughout the course allowed us to successfully design, analyze, and build the many circuits involved in the theremin's operation.